(19) Europäisches Patentamt
European Patent Office
Office européen des brevets

(11) Publication number: **0 661 860 A2**

(12) EUROPEAN PATENT APPLICATION

(21) Application number: **94309199.1**

(22) Date of filing: **09.12.94**

(51) Int. Cl.^{6}: **H04M 9/08**, H04M 1/60

(30) Priority: **29.12.93 US 175095**

(43) Date of publication of application:
05.07.95 Bulletin 95/27

(84) Designated Contracting States:
DE FR GB IT

(71) Applicant: **AT&T Corp.**
32 Avenue of the Americas
New York, NY 10013-2412 (US)

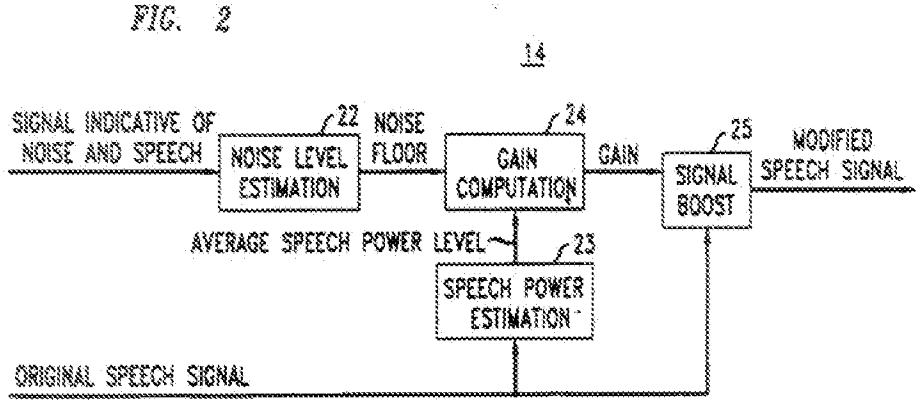
(72) Inventor: **Allen, Jonathan Brandon**
382 Forest Hill Way
Mountainside,
New Jersey 07092 (US)
Inventor: **Youtkus, Donald Joseph**
578 West Court
Scotch Plains,
New Jersey 07076 (US)

(74) Representative: **Buckley, Christopher Simon Thirsk et al**
AT&T (UK) LTD.,
AT&T Intellectual Property Division,
5 Mornington Road
Woodford Green,
Essex IG8 0TU (GB)

(54) Background noise compensation in a telephone network.

(57) An automated method for modifying a speech signal in a telephone network applies a gain factor (24) which is a function of the level of background noise (22) at a given destination, and transmits the modified speech signal (25) to the destination. The gain applied (by 25) may be a function of both the background noise level and the original speech signal. Either a linear or a non-linear (e.g., compressed) amplification of the original speech signal may be performed (in 25), where a compressed amplification results in the higher level portions of the speech signal being amplified by a smaller gain factor than lower level portions. The speech signal may be separated into a plurality of subbands, each resultant subband signal being individually modified in accordance with the present invention. In this case, each subband speech signal is amplified by a gain factor based on a corresponding subband noise signal, generated by separating the background noise signal into a corresponding plurality of subbands. The individual modified subband signals may then be combined to form the resultant modified speech signal.

FIG. 2



EP 0 661 860 A2

Cross-Reference to Related Application

The subject matter of this application is related to the U.S. Patent application of J. B. Allen and D. J. Youtkus entitled "Background Noise Compensation in a Telephone Set, Ser. No. 08/175038, filed on even date herewith and assigned to the assignee of the present invention.

Field of the Invention

The present invention relates generally to the field of telecommunications and specifically to the problem of using a telephone network to communicate with a party located in a noisy environment.

Background of the Invention

When a person communicates over a telephone network while located in a noisy environment, such as a noisy room, an airport, a car, a street corner or a restaurant, it can often be difficult to hear the person speaking at the other end (*i.e.*, the "far-end") of the connection over the background noise present at the listener's location (*i.e.*, the "near-end" or the "destination"). In some cases, due to the variability of human speech, the far-end speaker's voice is sometimes intelligible over the near-end background noise and sometimes unintelligible. Moreover, the noise level at the near-end may itself vary over time, making the far-end speaker's voice level at times adequate and at times inadequate.

Although terminal telephone equipment sometimes provides for control of the volume level of the telephone loudspeaker (*i.e.*, the earpiece), such control is often unavailable. Moreover, manual adjustment of a volume control by the listener is undesirable since, as the background noise level changes, the user will want to readjust the manual volume control in an attempt to maintain a preferred listening level. Generally, it is likely to be considered more desirable to provide an automatic (*i.e.*, adaptive) control mechanism, rather than requiring the listener first to determine the existence of the problem and then to take action by adjusting a manual volume control. One solution which attempts to address this problem has been proposed in U.S. Patent No. 4,829,565, issued on May 9, 1989 to Robert M. Goldberg, which discloses a telephone with an automated volume control whose gain is a function of the level of the background noise.

Summary of the Invention

We have recognized that the use of either conventional manual volume controls or an automatic mechanism such as that disclosed in the above-cited U.S. Patent No. 4,829,565 fails to adequately solve the background noise problem. In particular, these approaches fail to recognize the fact that by amplifying the signal which supplies the handset receiver (*i.e.*, the loudspeaker), the side tone is also amplified. (The side tone is a well-known feed-through effect in a telephone. A portion of the input signal from the handset transmitter -- *i.e.*, the microphone -- is mixed with the far-end speech signal received from the network. The resultant, combined signal is then supplied to the handset loudspeaker.) Since the side tone contains the background noise, itself, the background noise is, disadvantageously, amplified concurrently with the far-end speech signal whenever such a volume control (either manual or automatic) is used to amplify the signal which supplies the handset receiver. By amplifying both the speech signal and the noise together, the degrading effect of the noise can actually become worse because of the properties of the human ear.

Moreover, the use of either conventional manual volume controls or the automatic mechanism disclosed in the above-cited U.S. Patent No. 4,829,565 requires the use of specialized telephone terminal equipment. We have recognized that since there are millions of conventional telephone sets (without any such controls) presently in use, it is highly desirable that a mechanism which compensates for the presence of background noise be provided without requiring such specialized equipment.

In accordance with the present invention, background noise compensation is provided within a telephone network. In this manner, the far-end speech signal may, advantageously, be amplified as a function of the background noise without simultaneously amplifying the side tone. Moreover, the benefits of the invention are thereby provided to all users of the network, without any need to replace existing terminal telephone equipment with specialized equipment. As used herein, the term "telephone network" is intended to include conventional terrestrial telephone networks (local or long distance), wireless (including cellular) communication networks, radio transmission, satellite transmission, microwave transmission, fiber optic links, *etc.*, or any combination of any of these transmission networks.

Specifically, a modified speech signal is produced from an original speech signal within a telephone network destined for a given destination. The original speech signal is amplified by a gain factor to produce the modified speech signal. The gain factor is a function of a received signal indicative of the background noise at the destination. The modified signal is then communicated through the

network to the destination.

The gain factor may be a function of the level of the background noise, or it may be a function of both the level of the background noise and the level of the original (*i.e.*, the far-end) speech signal. The modified speech signal may comprise a linear amplification of the original speech signal or it may comprise an amplified and "compressed" version of the original speech signal. By "compressed" it is meant that the higher level portions of the original signal are amplified by a smaller gain factor than are the lower level portions.

In accordance with one illustrative embodiment, the original speech signal may be separated into a plurality of subbands, and each resultant subband signal may be individually modified (*e.g.*, amplified) in accordance with the technique of the present invention. In particular, these original subband speech signals may be amplified by a gain factor which is a function of a corresponding subband-noise- indicative signal. Such subband-noise-indicative signals may be generated by separating the signal indicative of the background noise into a corresponding plurality of subbands. The individual modified subband signals may then be combined to form the resultant modified speech signal.

Brief Description of the Drawings

Figure 1 shows a telephone network which includes a noise compensation system in accordance with an illustrative embodiment of the present invention.

Figure 2 shows a system-level diagram of a broadband-based illustrative embodiment of a noise compensation system in accordance with the present invention.

Figure 3 shows an illustrative implementation of the noise level estimation unit of the system of Figure 2.

Figure 4 shows an illustrative implementation of the gain computation unit of the system of Figure 2.

Figure 5 is a graph which shows a compressor gain which may be applied to the original speech signal by the signal boost unit of the system of Figure 2 applying compressed amplification.

Figure 6 is a graph of the corresponding transfer function for the illustrative signal boost unit which results from applying the gain shown in Figure 5.

Figure 7 shows an illustrative implementation of the signal boost unit of an embodiment of the system of Figure 2 applying a compressed amplification as shown in the graphs of Figures 5 and 6.

Figure 8 shows an alternative illustrative implementation of the gain computation unit of Figure 2 for use in an embodiment applying compressed amplification in an alternative manner.

Figure 9 shows a system-level diagram of a multiband-based illustrative embodiment of the present invention in which noise compensation is performed in individual subbands.

Detailed Description

Introduction

The present invention improves the signal-to-noise ratio (SNR) of a far-end speaker's speech in the near-end listener's ear when the near-end listener is using a telephone in a noisy environment. The level of the noise in the ear of the near-end listener can be estimated from the signal levels picked up by the transmitter (microphone) in the near-end listener's handset. Based on these levels, the original speech signal generated by the far-end speaker may be modified within the telephone network by being amplified by a variable gain factor so as to provide a more intelligible signal to the listener. This modification may advantageously also be a function of the level of the original speech signal itself. For example, the speech power level (*i.e.*, a "long-term" average level of the original speech signal) may be incorporated into the determination of the gain factor. In this manner, relatively quiet signals may be boosted (*i.e.*, amplified) by a larger gain factor than relatively loud signals.

Moreover, the modification of the speech signal may comprise either a linear amplification or a nonlinear, (illustratively) compressed, amplification. Compressed amplification, in particular, boosts loud portions of the original speech signal by a lesser amount (*i.e.*, with a smaller gain factor) than quiet portions. Thus, it is possible in this manner to, on a short-term basis, boost the signals which fall below the background noise level without boosting the signals which are already significantly above the background noise level. Simple linear amplification, by contrast, boosts all signal levels by an equal amount. When used to boost low-level signals above the background noise, linear amplification can in some circumstances result in distortion, since the higher level signals (already above the noise) could receive excessive amplification.

Figure 1 shows a telephone network which includes a noise compensation system embodying the principles of the present invention. A far-end speaker provides an original speech signal through microphone **11m** (of telephone handset **11h**) of conventional far-end telephone **11**. (Telephone handset **11h** also includes loudspeaker **11s** and telephone **11** also includes deskset **11d**.) This original speech signal, after being processed by telephone

network **12** in accordance with the principles of the present invention, is transmitted to a near-end listener using conventional near-end telephone **13**. Telephone **13** comprises deskset **13d** and handset **13h**. Loudspeaker **17** represents the presence of background noise at the near-end location.

Noise compensation system **14**, contained within telephone network **12**, receives a noise-indicative signal from near-end telephone **13** (provided by microphone **13m** contained in handset **13h**). This noise-indicative signal includes the background noise in the near-end environment, and may further include any speech provided to telephone **13** by the near-end listener. Noise compensation system **14** also receives the original speech signal from the far-end speaker (provided by far-end telephone **11**).

In summary, noise compensation system **14** first determines the level of background noise by recognizing and removing any (near-end) speech component from the noise-indicative signal. Next, noise compensation system**14** boosts the original speech signal based on the determined background noise level to produce a modified speech signal. The modified speech signal is then transmitted to near-end telephone **13** for broadcast through loudspeaker **13s** contained in handset **13h**. By including noise compensation system **14** within telephone network **12**, the benefits of noise compensation may be obtained with use of conventional terminal telephone equipment.

Figure 1 also shows telephone switches **15f** and **15n**, which connect to far-end telephone **11** and near-end telephone **13**, respectively. Switches **15f** and **15n** comprise conventional telephone switching devices. Figure 1 further shows conventional hybrids **16f** and **16n**, which comprise conventional circuits for converting between standard two-wire and four-wire telephone lines.

An Illustrative Broadband Implementation with Linear Amplification

Figure 2 shows a system-level diagram of a broadband-based illustrative embodiment of noise compensation system **14**. Inputs to the system include the original speech signal and the noise-indicative signal, which may further include speech provided by the near-end listener. The system produces a modified speech signal for improved intelligibility as output. All of the signals described with reference to the illustrative embodiment present herein are presumed to be in digital form.

Based on the noise-indicative signal, noise level estimation **22** determines the "noise floor" and outputs a signal representing that value. In particular, this signal represents the noise level over a first predetermined period of time. By setting this first predetermined period to a relatively short value (*e.g.*, 250 milliseconds or less), the determined noise floor will substantially follow changing levels of background noise in the near-end environment. Specifically, the noise floor signal represents a short-term (*e.g.*, 250 milliseconds) minimum value of an "exponentionally mapped past average" signal, and can be generated using known techniques. An illustrative implementation of noise level estimation **22** is shown in Figure 3 and described below.

Gain computation **24** produces a gain signal, GAIN, whose value is proportional to the noise floor signal and inversely proportional to an average speech power level signal. This gain signal represents a gain factor (*i.e.*, a multiplicative factor) by which the original speech signal may be amplified. The average speech power level signal is generated by speech power estimation **23**, and represents the average level of the original speech signal over a second predetermined period of time. That is, the average speech power level measures the "energy" level of the speech signal. Providing such a gain dependence on the far-end speech level allows relatively quiet calls to receive a sufficient boost for a given background noise level, while preventing loud calls from being over-boosted. By setting the second predetermined period to a relatively long value (*e.g.*, one second), it can more readily be determined whether the current far-end speech comprises a loud or soft segment of the call. Thus, the average speech power level signal represents a long-term average level. Speech power estimation **23** may be implemented by conventional signal energy estimation techniques. An illustrative implementation of gain computation **24** is shown in Figure 4 and described below.

The gain signal and the original speech signal are provided to signal boost **25** which produces the modified speech signal. Where only linear amplification is desired, signal boost **25** may comprise a conventional amplifier (*i.e.*, a multiplier). In this case, the original speech signal is amplified by a gain factor equal to the value of the gain signal, GAIN. Where, on the other hand, compressed amplification is desired, signal boost **25** may comprise circuitry (or procedural code) which amplifies the original speech signal by a gain factor less than or equal to the value of the gain signal, wherein the gain factor further depends on the level of the original speech signal itself. That is, the gain signal, GAIN, represents the maximum gain which will be applied by the "compressor." An illustrative implementation of signal boost **25** providing compression is shown in Figure 7 and described below.

Figure 3 shows an illustrative implementation of noise level estimation **22** of the system of Figure 2. First, high pass filter (HPF) **31** removes DC from

the input signal. It may be conventionally implemented as a first order recursive digital filter having a cutoff frequency of, for example, 20 Hz, and may be based on a standard telephony sampling frequency of 8 kHz. Absolute value block (ABS)**32** computes the magnitude of the sample and is also of conventional design. Low pass filter (LPF) **33** computes the exponentially mapped past average (EMP). As described above, the exponentially mapped past average comprises an exponentially weighted average value of the noise level. Low pass filter **33** is also of conventional design and may illustratively be implemented as a first order recursive digital filter having the transfer function $y(n) = (1-\beta)\, x(n) + \beta\, y(n-1)$, where $\beta = e^{-T/\tau}$, with T a sampling period and τ a time constant. Illustratively, T = 0.125 ms and τ = 16 ms.

Minimum sample latch (MIN) **34** stores the minimum value of EMP over the first predetermined time period (*e.g.*, 250 milliseconds). The output signal of latch **34**, MEMP, therefore represents the short-term minimum of the exponentionally mapped past average, and thus represents the short-term minimum value of the averaged noise-indicative signal. This signal is subsequently used to represent the noise floor over which far-end speech should be boosted. In a corresponding manner, maximum sample latch (MAX) **35** stores the maximum value of EMP over the same predetermined period. The output signal of latch **35**, PEMP, therefore represents the short-term peak of the exponentionally mapped past average, and thus represents the short-term peak value of the averaged noise-indicative signal. Latches **34** and **35** may be implemented by conventional digital comparators, selectors and storage devices, with the storage devices reset at the start of each cycle of the predetermined time period.

Speech detector and noise floor estimator **36** generates the noise floor signal output based on signals MEMP and PEMP. Specifically, it performs two functions. First, it is determined whether the noise-indicative signal presently includes only noise or whether it presently includes speech as well. This question may be resolved by conventional techniques, such as those used in the implementation of conventional speakerphones. For example, the quotient of PEMP (representing the short-term peak value of the noise-indicative signal) divided by MEMP (representing the short-term minimum value of the noise-indicative signal) may be compared with a predetermined threshold. The larger this quotient, the larger the variability in the level of the input signal. If the level of the input signal is sufficiently variable within the first predetermined time period, it is presumed that speech is present. (Note that the variation in signal level of speech typically exceeds that of background noise.)

Second, speech detector and noise floor estimator **36** sets the output noise floor signal to a value which represents the estimated level of the noise floor. If it is determined that speech is not present, the noise floor signal is set to MEMP, the short-term minimum value of the noise-indicative signal. Otherwise, the noise floor signal remains unchanged -- that is, the previous value is maintained. In this manner, when the presence of speech makes it difficult to determine the actual present level of background noise, it is presumed that the noise level has not changed since the previous period.

In one alternative embodiment, the value of PEMP alone may be compared with a predetermined threshold (rather than using the quotient of PEMP divided by MEMP), since speech is generally of a significantly higher intensity than is background noise. And in a second alternative embodiment, speech detection may be bypassed altogether, on the assumption that the far-end speaker will not be speaking at the same time that the near-end listener is speaking. In other words, we may not care what the "noise floor" is determined to be during periods when the near-end listener is speaking. In this second alternative embodiment, maximum sample latch **35** and speech detector and noise floor estimator **36** may be removed from noise level estimation **22** of Figure 3, and the output of minimum sample latch **34** (*i.e.*, signal MEMP) may be used directly as the noise floor signal output of noise level estimation **22**.

Figure 4 shows an illustrative implementation of gain computation **24** of the system of Figure 2. The gain signal is generated based on the noise floor signal from noise level estimation **22** and on the average speech power level signal from speech power estimation **23**. Specifically, the computed gain is advantageously proportional to the noise floor and inversely proportional to the average speech power level. Moreover, the gain is never less than one (*i.e.*, the original speech signal is never attenuated) nor is it ever more than a maximum specified value.

First, amplifier **41** multiplies the noise floor by a noise scale factor. This noise scale factor is set to an appropriate value so that the output signal of amplifier **41**, which is representative of a gain factor, is of the appropriate magnitude. In particular, the noise scale factor acts as a "sensitivity" control -- a smaller scale factor will result in more gain being applied for a given level of background noise. The magnitude of this signal may be advantageously set to that gain factor which will boost the lowest far-end speech levels by an appropriate amount to overcome the noise level. For example, the noise scale factor may illustratively be set to a

fractional value between zero and one, such as 0.4.

Next, minimizer (MIN) **42** compares the gain factor output by amplifier **41** to a maximum permitted gain factor to ensure that the system does not attempt to apply an excessive gain factor to the original speech signal. For example, the maximum permitted gain factor may illustratively be set to 5.6 (*i.e.*, 15 dB). Maximizer (MAX) **43** then ensures that the resultant gain factor is in no case less than one, so that the original speech signal is never attenuated.

Divider **44** and minimizer (MIN) **45** determine an additional multiplicative factor to be incorporated in the gain computation so that the resultant gain will be inversely proportional to the average speech power level as provided by speech power estimation **23**. Divider **44** computes the quotient of a minimum far-end speech level divided by the average speech power level for use as this additional multiplicative factor. The minimum speech level represents the minimum level which is to be considered actual far-end speech, as distinguished from mere background noise during a period of silence by the far-end speaker. For example, the minimum speech level may illustratively be set to a value representing -30 dBm. Minimizer **45** then ensures that this multiplicative factor does not exceed one. In this manner, the gain factor is not increased as the far-end speech level goes below the minimum, so that far-end background noise is not over-boosted (*i.e.*, not boosted more than the quietest speech).

Amplifier **46** multiplies the gain factor generated by amplifier **41** (through minimizer **42** and maximizer **43**) by the additional multiplicative factor from divider **44** (through minimizer **45**). Finally, maximizer (MAX) **47** ensures that the final gain factor is not less than one, so that the original speech signal is never attenuated. Thus, the resultant gain factor, GAIN, is proportional to the noise floor level and inversely proportional to the average speech power level, but neither less than one nor more than the specified maximum.

An Illustrative Broadband Implementation with Compressed Amplification

As described above, the technique of compressed amplification results in the application of more gain to lower energy signals than to higher energy signals. This helps to compensate for the listener's reduced dynamic range of hearing and undue growth of loudness which results from the presence of surrounding noise. Since lower energy signals tend to be masked by noise more than higher energy signals, the higher energy signals require less amplification. Moreover, this compression avoids distorting the speech by avoiding over-amplification of the high energy signals. Thus, the speech intelligibility is increased without the unwanted side effect of over-amplifying those sounds which are already sufficiently loud.

Figure 5 is a graph which shows a compressor gain which may be applied to the original speech signal by the signal boost unit of an illustrative embodiment of the system of Figure 2 applying compressed amplification. Figure 6 is a graph which shows the corresponding transfer function for the illustrative signal boost unit which results from applying the gain shown in Figure 5. As shown, the gain (in decibels, or dB) to be applied varies from GL, a predetermined "low-level" gain which is applied to the lowest energy signals, down through GH, a "high-level" gain, to no gain at all (*i.e.*, 0 dB) at the highest energy signals. The low-level gain, GL, may be based on the output of gain computation **24**, GAIN, as shown in Figure 4 and described above. In particular, where GAIN reflects a maximum gain factor and GL reflects a gain in decibels, it can be readily seen that GL = 20 log (GAIN). Note from the graphs of Figures 5 and 6 that the gain advantageously remains non-negative, thus ensuring that the signal is never attenuated.

The compressor "breakpoint," BK, is an original speech signal level threshold below which the gain applied remains constant. That is, signals below BK receive a linear boost while only those above BK are in fact compressed. By keeping the gain applied constant below this threshold, very low level signals, which likely represent only background noise at the far end (rather than actual far-end speech), will not be excessively amplified (*i.e.*, will not be boosted more than the lowest level speech signals), while low level speech signals will still receive sufficient boost. P represents a point at which a high-level gain, GH, may be defined. Both the compressor breakpoint BK and the point P may be advantageously chosen so that most of the dynamic range of the original speech signal falls between BK and P. Thus, the low-level gain GL will be applied to the lowest level speech signals, while the high-level gain, GH, will be applied to the highest level speech signals. For example, BK may be set at the minimum level which represents actual speech (as opposed to far-end background noise). P, for example, may be set at a speech level which is exceeded only 10% of the time. Alternatively, since speech typically has an energy distribution that ranges over about 30 dB, either BK or P may be chosen as indicated above, and then the other parameter may be set 30 dB higher or lower, respectively.

Figure 7 shows an illustrative implementation of the signal boost unit of the embodiment of the system of Figure 2 applying a compressed amplification as shown in the graphs of Figures 5 and

6. The illustrative implementation comprises absolute value block (ABS) **50**, peak detector **51**, logarithm block (LOG) **52**, multiplier **53**, adder **54**, minimizer (MIN) **55**, adder **56**, maximizer (MAX) **57**, exponentiator (EXP) **58** and multiplier **59**. As can be seen from the presence of logarithm block **52** and exponentiator **58**, the computation of the compressed gain is primarily performed in the logarithmic domain. All of the individual components are of conventional design.

Specifically, absolute value block **50** computes the magnitude of the sample. Peak detector **51** controls the attack and release times of the compressor. For example, peak detector **51** may be advantageously designed so as to provide instantaneous attack but syllabic release. An instantaneous attack time enables the compressor gain to be reduced instantaneously if the input signal level suddenly rises. Therefore, sudden, loud noises are prevented from being over-amplified, thus avoiding causing pain or injury to the listener's ear. The compressor gain increases, however, at a rate dependent on the release time constant. The release time constant may be set, for example, to 16 milliseconds (or less) to respond to the fast energy changes associated with the phonemes of spoken language. Specifically, if $x(n)$ represents the n'th input sample to peak detector **51** and $y(n)$ represents the n'th output sample therefrom, peak detector **51** may be implemented by setting $y(n) = x(n)$ if $x(n) > y(n-1)$, and otherwise setting $y(n) = \beta\, y(n-1)$, where $\beta = e^{-T/\tau}$, with T set equal to the sampling period (*e.g.*, 0.125 milliseconds for telephony) and τ set equal to the release time constant (*e.g.*, 25 milliseconds).

Logarithm block **52** converts the output signal of peak detector **51** into the logarithmic domain by taking the logarithm of the digital sample. Multiplier **53**, adder **54** and minimizer **55** compute the relative reduction in gain which is to result from the compression. That is, the amount by which the resultant gain will be reduced from the low-level gain, GL, (which represents the maximum gain) is calculated by these components. Specifically, multiplier **53** multiplies the signal by the amount (k-1), where k is the reciprocal of the "compression ratio." The compression ratio, CR, represents the slope of the compressor gain curve as shown in Figure 6, and may be easily calculated from the parameters BK, P, GL and GH (as defined above) as $CR = 1/k = (P - BK)/(P - BK + GH - GL)$. Adder **54** then adds the (negative) amount $-(k-1)\log(bk)$ to the result from multiplier **53**, where bk is the compressor breakpoint (*i.e.*, BK) expressed as an absolute level on a linear scale. For example, if the speech signal magnitudes are in the range [0,R] on a linear scale and it is desired that the compressor breakpoint be placed a predetermined amount x dB down from R, then $bk = R \times 10^{(-x/20)}$. Minimizer **55** limits the result of the above computation to a value less than or equal to zero so that the final resultant compressed gain will never exceed the low-level gain, GL.

Adder **56** adds in the amount gl, which is the logarithm of the gain which is introduced by the compressor at all levels less than bk (*i.e.*, the low-level gain). Thus, $gl = \log(GAIN) = GL / 20$. Maximizer **57** ensures that the final result (as computed in the logarithmic domain) remains greater than or equal to zero to ensure that the original speech signal is never attenuated. Exponentiator **58** converts the computed compressed gain back out of the logarithmic domain to produce the final gain factor (*i.e.*, the compressed gain). Finally, multiplier **59** applies this (multiplicative) gain factor to the original speech signal to produce the modified speech signal.

An Alternative Illustrative Implementation of Compressed Amplification

Figure 8 shows an alternative illustrative implementation of the gain computation unit of Figure 2 for applying compressed amplification in a different manner than that described above. In gain computation 24' shown herein, the low-level gain, GL, of the compressor of signal boost **25** is varied only as a function of the background noise level (and not based on the average speech power level), while the high-level gain, GH, is varied as a function of the average speech power level. That is, the low-level gain is proportional (only) to the noise floor, and the high-level gain is inversely proportional (only) to the average speech power level. Thus, gain computation 24' produces an output (GAIN) comprising two "independent" gain factors, both of which are supplied to signal boost **25**.

For example, if P is chosen to be set at a speech level which is exceeded only 10% of the time as suggested above, the result of this alternative implementation is that the effect of varying the low-level gain becomes essentially orthogonal to the effect of varying the high-level gain. In particular, varying the low-level gain will affect the intelligibility of the speech but the loudness will be relatively unaffected if the high-level gain remains constant. On the other hand, varying the high-level gain will affect the loudness of the speech but the intelligibility will be relatively unaffected if the low-level gain remains constant. Thus, the low-level gain becomes an intelligibility "control" and the high-level gain becomes a loudness "control." Advantageously, therefore, the illustrative implementation described herein increases the low-level gain as the background noise increases, while it increases the high-level gain as the far-end speech

level decreases.

Specifically, in the alternative implementation of Figure 8, amplifier **41**, minimizer (MIN) **42** and maximizer (MAX) **43** produce a gain factor proportional to the noise floor in an analogous manner to the corresponding components of the implementation shown in Figure 4. The same parameters -- a noise scale factor and a maximum permitted gain factor -- are employed in the same manner. The resultant signal in this case, however, is the final low-level gain factor to be provided to the compressor of signal boost **25**.

Divider **44** and minimizer (MIN) **45** determine an alternative gain factor (inversely proportional to the average speech power level), also in an analogous manner to the corresponding components of the implementation shown in Figure 4. Multiplier **48** then multiplies this factor (which is less than or equal to one) by a parameter representing the maximum permitted high-level gain factor to produce the high-level gain factor to be provided to the compressor of signal boost **25**. For example, the maximum permitted high-level gain factor may advantageously be set to the low-level gain factor. Maximizer **49**, like maximizer **43**, ensures that the resultant gain factor is at least one, so that the original speech signal is never attenuated.

With the resultant gain factors as produced by gain computation 24', signal boost **25** may be implemented as shown in Figure 7 and described above. In particular, the compression ratio, CR, may be readily computed as described above based on the low-level and high-level gain factors generated by gain computation 24'. The compressed gain may then be computed based on the values of k (1/CR), bk and gl (based in turn on the low-level gain factor) as described above.

An Illustrative Multiband Implementation

Figure 9 shows a system-level diagram of a multiband-based illustrative embodiment of the present invention in which noise compensation is performed in individual (frequency) subbands. By performing noise compensation independently in distinct subbands, the noise energy in one frequency band will not affect the gain applied to the original speech signal at other frequencies. For example, high energy, low frequency components in the original speech signal will advantageously not affect the gain applied to the high frequency components of the signal. In general, multiband-based noise compensation permits better adaptation to the spectral characteristics of the background noise.

The structure and operation of the illustrative multiband system corresponds generally to that of the broadband system of Figure 2. However, each of the processes performed by the broadband system of Figure 2 is performed by the multiband system of Figure 9 in a plurality of independent subbands. In particular, each of the four components shown in Figure 2 may be replaced by a plurality of corresponding "copies" of the given component, each of which operates on one of the n subbands into which each of the input signals is separated. Since subband-based processing of speech and audio signals is well known, the following description provides an overview of the multiband implementation of Figure 9.

Specifically, multiband noise compensation system 14' comprises analysis filter banks **61** and **62**, noise level estimation 22', speech power estimation 23' , gain computation 24', and signal boost 25' and adder **63**. (Units which correspond to those of the broadband system of Figure 2 have been assigned the same numbers with an added "prime" mark.) Each of the two input signals -- the noise-indicative signal and the original speech signal -- are separated into a corresponding set of n subband signals by analysis filter banks **61** and **62** in a conventional manner. Advantageously, these two filter banks are identical so that the two signals are separated into corresponding sets of subband signals having exactly the same frequency band structure.

Noise level estimation 22' comprises subband noise level estimation **22-1**,...**22-n**; speech power estimation 23' comprises subband speech power estimation **23-1**,...**23-n**; gain computation 24' comprises subband gain computation **24-1**,...**24-n**; and signal boost 25' comprises subband signal boost **25-1**,...**25-n**. Each corresponding set of components **22-i**, **23-i**, **24-i** and **25-i** (corresponding to the i'th subband) have a corresponding internal structure and operate in an analogous manner to components **22**, **23**, **24** and **25** of broadband noise compensation system **14** of Figure 2. After the speech signal as divided into subbands has been appropriately modified in each of these subbands (by subband signal boost **25-1**,...**25-n**), adder **63** combines the resultant modified subband speech signals to produce the final modified speech signal for use at the destination. Adder **63** is of conventional design.

In an alternative multiband embodiment, speech power estimation is not performed in subbands. In this case, speech power estimation **23** of the broadband system of Figure 2 may be used in place of speech power estimation 23' , providing its output signal (average speech power level) to each of the subband gain computation components (**24-1**,...**24-n**). That is, this alternate embodiment provides gain factors in each subband which are inversely proportional to the overall speech power level of the original speech signal as a whole,

rather than to the power level in each subband individually.

Although the individual subband components of multiband noise compensation system 14' correspond to the components of noise compensation system **14**, the various parameters (*e.g.*, the noise scale factor, the maximum permitted gain factor, the minimum speech level, *etc.*) described in connection with noise compensation system **14** above may be advantageously assigned different values in the different subband implementations. For example, in a multiband compression system, the release time of peak detector **51** in a higher frequency band may be advantageously set lower than the release time for a corresponding peak detector in a lower frequency band.

For clarity of explanation, the illustrative embodiment of the present invention is presented as comprising individual functional blocks. The functions these blocks represent may be provided through the use of either shared or dedicated hardware, including, but not limited to, hardware capable of executing software. For example, the functions of processors presented in the various figures may be provided by a single shared processor. (Use of the term "processor" should not be construed to refer exclusively to hardware capable of executing software.)

Illustrative embodiments may comprise digital signal processor (DSP) hardware, read-only memory (ROM) for storing software performing the operations discussed below, and random access memory (RAM) for storing DSP results. Very large scale integration (VLSI) hardware embodiments, as well as custom VLSI circuitry in combination with a general purpose DSP circuit, may also be provided.

Although a number of specific embodiments of this invention have been shown and described herein, it is to be understood that these embodiments are merely illustrative of the many possible specific arrangements which can be devised in application of the principles of the invention. Numerous and varied other arrangements can be devised in accordance with these principles by those of ordinary skill in the art without departing from the spirit and scope of the invention.

Claims

1. A method of processing an original speech signal in a telephone network to produce a modified speech signal, the modified speech signal for use at a destination having background noise thereat, the method comprising the steps of:

   receiving a signal indicative of the background noise at the destination;

   applying a gain to the original speech signal to produce the modified speech signal, wherein the gain is a function of the signal indicative of the background noise; and

   transmitting the modified speech signal through the telephone network to the destination.

2. A method of processing an original speech signal in a telephone network to produce a modified speech signal, the modified speech signal for use at a destination having background noise thereat, the method comprising the steps of:

   receiving a signal indicative of the background noise at the destination;

   separating the original speech signal into a plurality of original subband speech signals;

   separating the signal indicative of the background noise into a plurality of subband-noise-indicative signals corresponding to the plurality of original subband speech signals;

   applying a corresponding subband gain to each original subband speech signal to produce a corresponding plurality of modified subband speech signals, wherein each subband gain is a function of the corresponding subband-noise-indicative signal;

   combining the plurality of modified subband speech signals to produce the modified speech signal; and

   transmitting the modified speech signal through the telephone network to the destination.

3. A method for use in providing a telephone network service comprising the steps of:

   receiving an original speech signal for transmission to a destination having background noise thereat;

   receiving a signal indicative of the background noise at the destination;

   applying a gain to the original speech signal to produce a modified speech signal, wherein the gain is a function of the signal indicative of the background noise; and

   transmitting the modified speech signal to the destination.

4. The method of claim 3 further comprising the step of measuring a level of the signal indicative of the background noise over a first predetermined time period, wherein the gain is a function of said measured level.

5. The method of claim 3 wherein the gain is a further function of the original speech signal.

6. The method of claim 5 further comprising the step of determining an energy level of the original speech signal measured over a second predetermined time period, wherein the gain is a further function of said energy level.

7. The method of claim 5 further comprising the step of determining the gain, wherein the gain is a further function of a level of the original speech signal, and wherein the gain applied to the original speech signal when it is at a first level is greater than the gain applied to the original speech signal when it is at a second level greater than said first level.

8. The method of claim 3 wherein the signal indicative of the background noise comprises a signal indicative of both the background noise and speech, and wherein the step of applying the gain includes the step of determining when said signal indicative of both the background noise and speech does not include speech and determining the gain at such times.

9. A method of processing an original speech signal in a telephone network to produce a modified speech signal, the modified speech signal for use by a telephone set at a destination having background noise thereat, the telephone set including means for receiving the modified speech signal from the telephone network and means for adding a side tone to the received signal, the method comprising the steps of:

receiving a signal indicative of the background noise at the destination;

applying a gain to the original speech signal to produce the modified speech signal, wherein the gain is a function of the signal indicative of the background noise; and

transmitting the modified speech signal through the telephone network to the telephone set at the destination,

whereby the gain is applied to the original speech signal to produce the modified speech signal before the side tone is added thereto.

10. The method of claim 1, or 9 wherein the gain is a function of a level of the signal indicative of the background noise measured over a first predetermined time period.

11. The method of claim 1 or 9 wherein the gain is a further function of the original speech signal.

12. The method of claim 3 or 11 wherein the gain is a further function of an energy level of the original speech signal measured over a second predetermined time period.

13. The method of claim 1 or 9 wherein the gain is a further function of a level of the original speech signal and wherein the gain applied to the original speech signal when it is at a first level is greater than the gain applied to the original speech signal when it is at a second level greater than said first level.

14. The method of claim 1 or 9 wherein the signal indicative of the background noise comprises a signal indicative of both the background noise and speech, and wherein the step of applying the gain includes the step of determining when said signal indicative of both the background noise and speech does not include speech and determining the gain at such times.

15. A method of processing an original speech signal in a telephone network to produce a modified speech signal, the modified speech signal for use by a telephone set at a destination having background noise thereat, the telephone set including means for receiving the modified speech signal from the telephone network and means for adding a side tone to the received signal, the method comprising the steps of:

receiving a signal indicative of the background noise at the destination;

separating the original speech signal into a plurality of original subband speech signals;

separating the signal indicative of the background noise into a plurality of subband-noise-indicative signals corresponding to the plurality of original subband speech signals;

applying a corresponding subband gain to each original subband speech signal to produce a corresponding plurality of modified subband speech signals, wherein each subband gain is a function of the corresponding subband-noise-indicative signal;

combining the plurality of modified subband speech signals to produce the modified speech signal; and

transmitting the modified speech signal through the telephone network to the telephone set at the destination,

whereby the subband gains are applied to the corresponding original subband speech signals to produce the corresponding modified subband speech signals before the side tone is added to the modified speech signal.

16. The method of claim 7 or 15 wherein each subband gain is a function of a level of the corresponding subband-noise-indicative signal

measured over a first predetermined time period.

17. The method of claim 7 or 15 wherein each subband gain is a further function of the corresponding original subband speech signal.

18. The method of claim 8 or 17 wherein each subband gain is a further function of an energy level of the corresponding original subband speech signal measured over a second predetermined time period.

19. The method of claim 7 or 15 wherein each subband gain is a further function of the original speech signal.

20. The method of claim 11 or 19 wherein each subband gain is a further function of an energy level of the original speech signal measured over a second predetermined time period.

21. The method of claim 7 or 15 wherein each subband gain is a further function of a level of the corresponding original subband speech signal and wherein the subband gain applied to the original subband speech signal when it is at a first level is greater than the subband gain applied to the original subband speech signal when it is at a second level greater than said first level.

22. The method of claim 7 or 15 wherein the signal indicative of the background noise comprises a signal indicative of both the background noise and speech, and wherein the step of applying the subband gains includes the step of determining when said signal indicative of both the background noise and speech does not include speech and determining the subband gains at such times.

FIG. 1

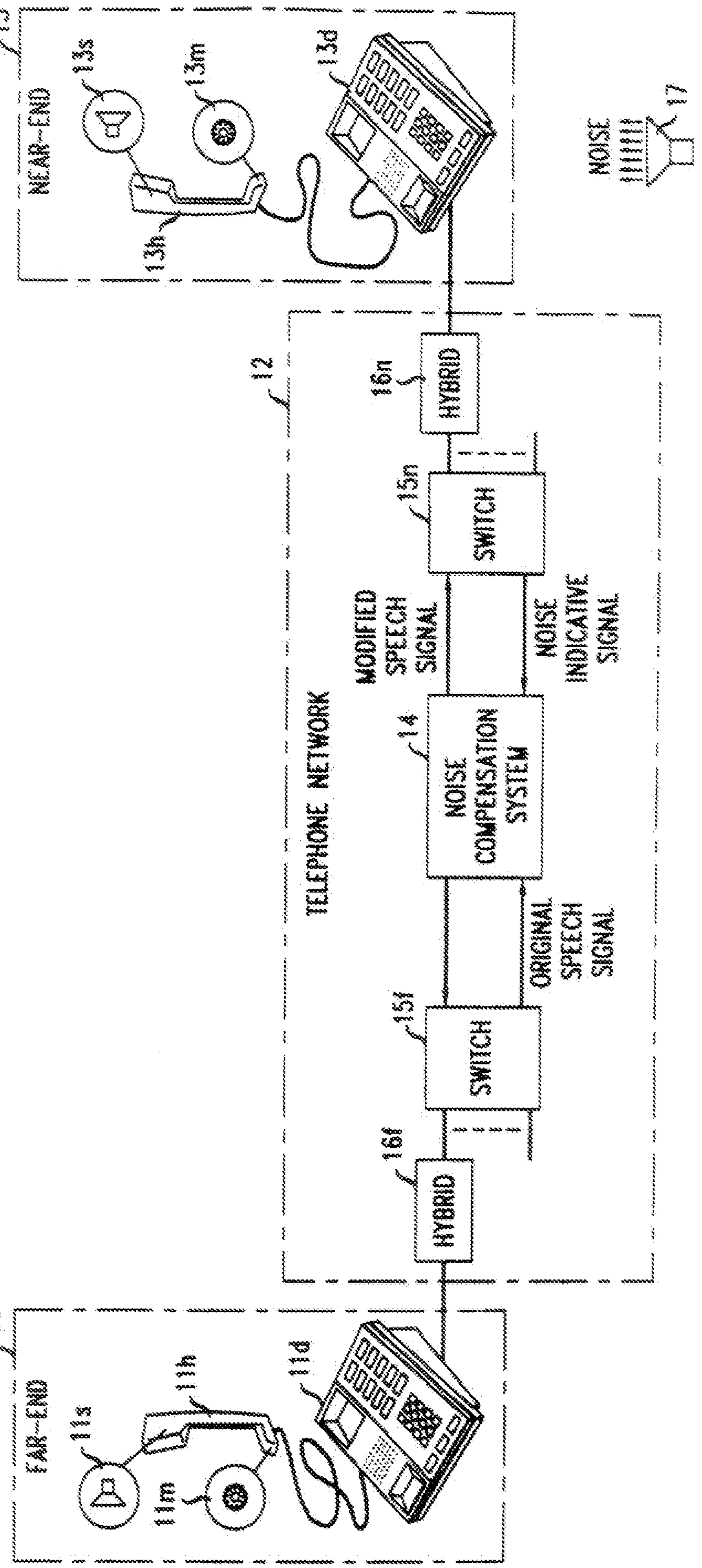

FAR-END
11
11s
11h
11m
11d
HYBRID
16f
SWITCH
15f
ORIGINAL SPEECH SIGNAL
TELEPHONE NETWORK
12
NOISE COMPENSATION SYSTEM
14
MODIFIED SPEECH SIGNAL
NOISE INDICATIVE SIGNAL
SWITCH
15n
HYBRID
16n
NEAR-END
13
13s
13m
13h
13d
NOISE
17

FIG. 2

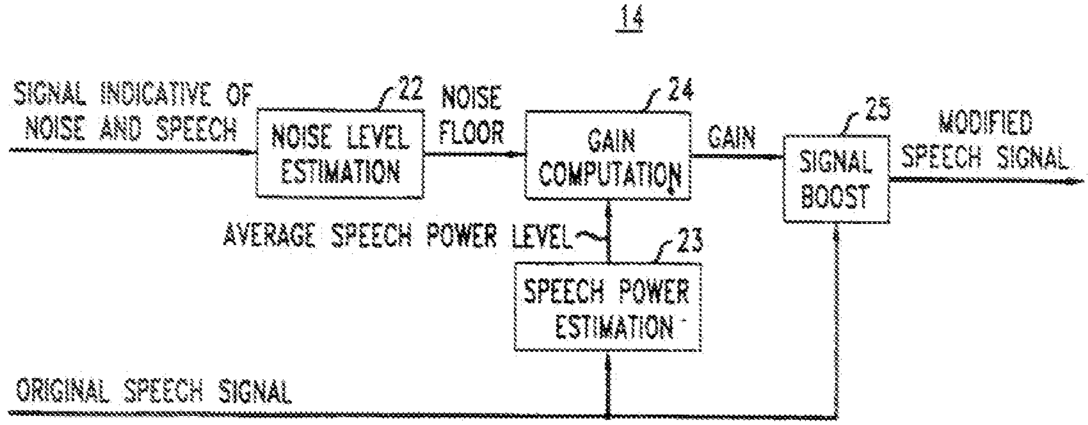


FIG. 3

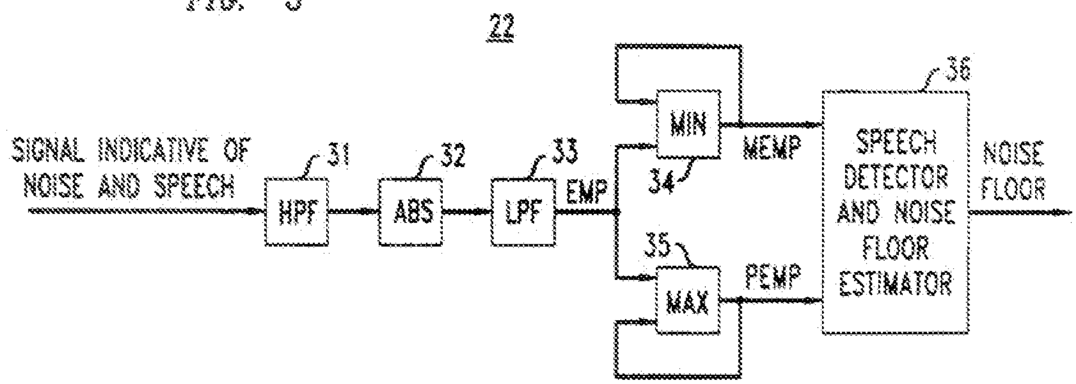


FIG. 4

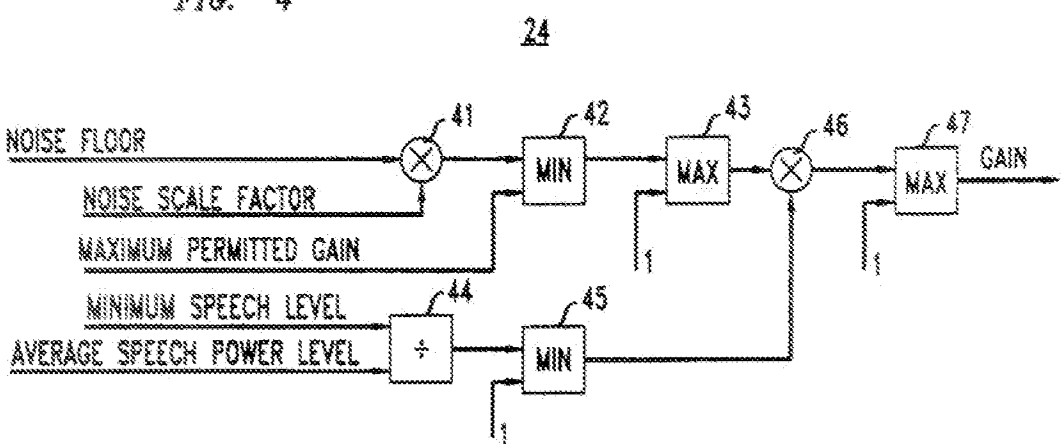


FIG. 5

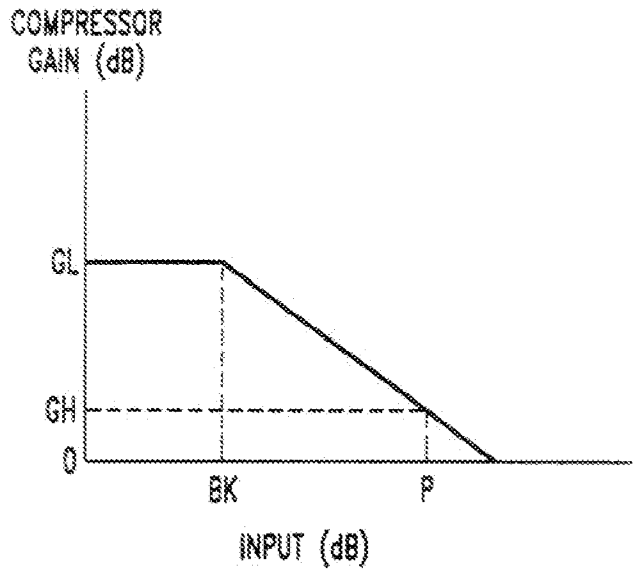

COMPRESSOR
GAIN (dB)
GL
GH
0
BK
P
INPUT (dB)

FIG. 6

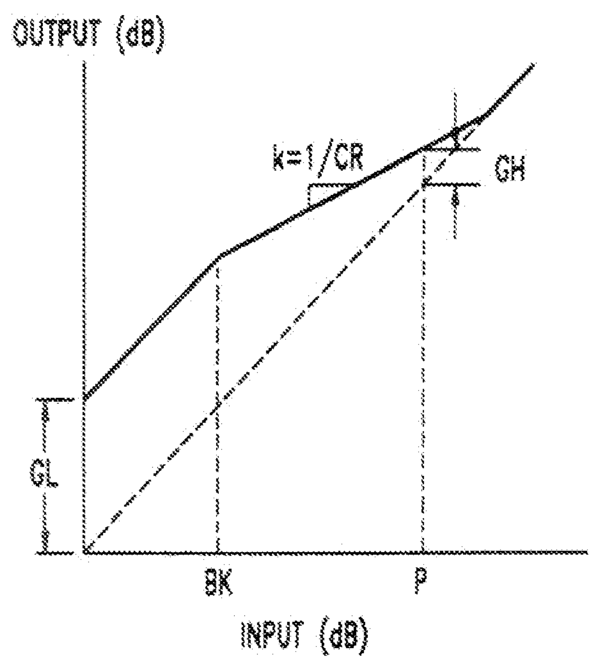

OUTPUT (dB)
k=1/CR
GH
GL
BK
P
INPUT (dB)

FIG. 7

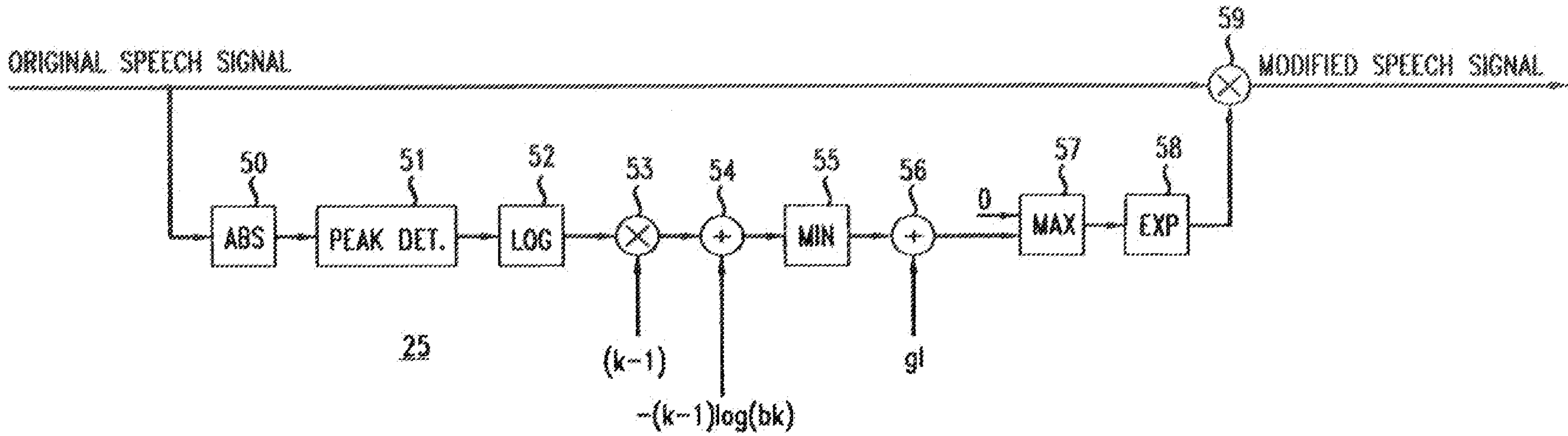

ORIGINAL SPEECH SIGNAL
MODIFIED SPEECH SIGNAL
59
50
ABS
51
PEAK DET.
52
LOG
53
54
55
MIN
56
57
MAX
58
EXP
0
25
(k-1)
-(k-1)log(bk)
gl

FIG. 8

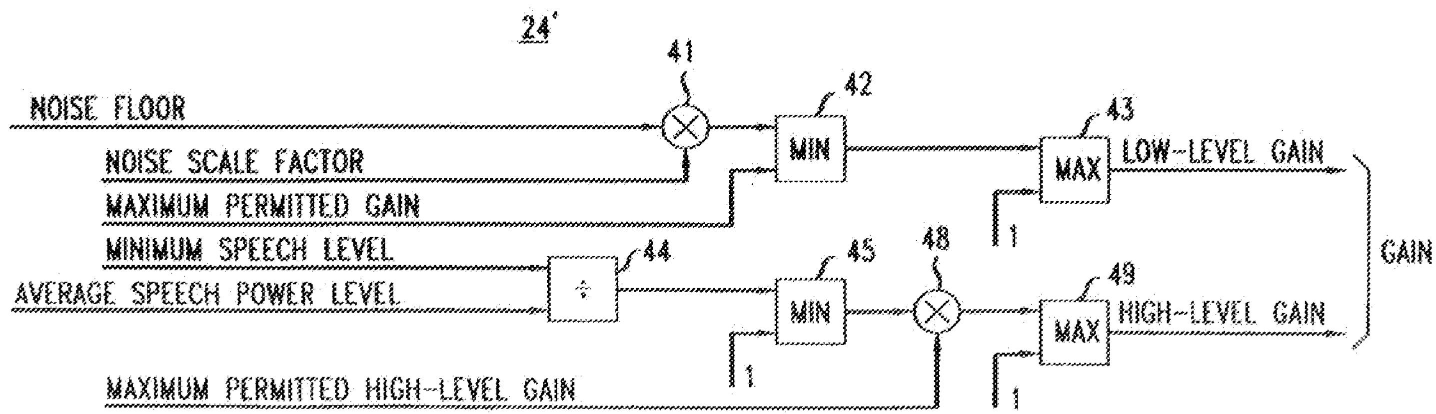
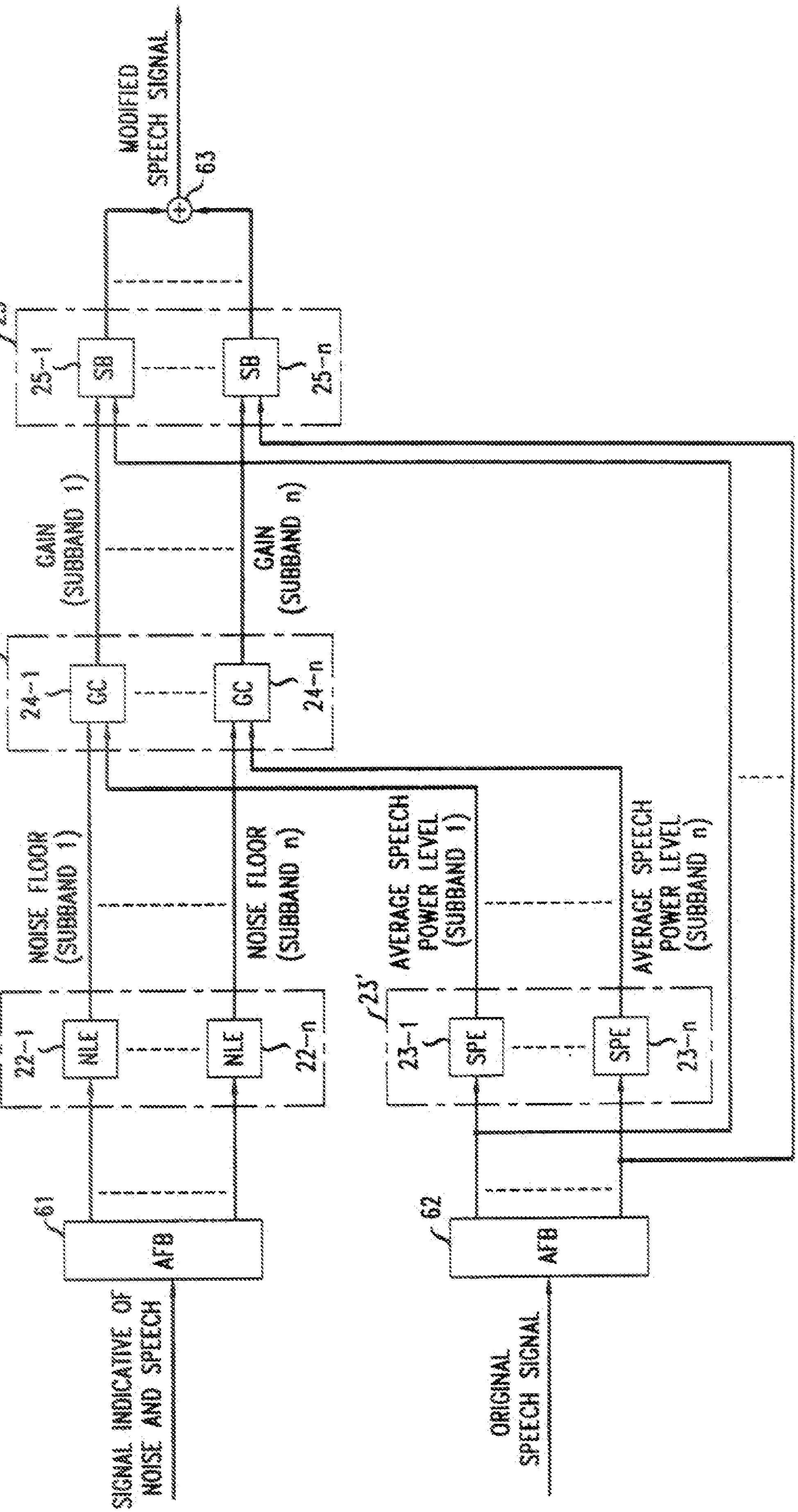

24'
NOISE FLOOR
NOISE SCALE FACTOR
MAXIMUM PERMITTED GAIN
MINIMUM SPEECH LEVEL
AVERAGE SPEECH POWER LEVEL
MAXIMUM PERMITTED HIGH-LEVEL GAIN
41
42
MIN
43
MAX
LOW-LEVEL GAIN
44
÷
45
MIN
48
49
MAX
HIGH-LEVEL GAIN
1
1
1
GAIN

FIG. 9


SIGNAL INDICATIVE OF NOISE AND SPEECH
ORIGINAL SPEECH SIGNAL
61
AFB
62
AFB
22'
22–1
NLE
NLE
22–n
23'
23–1
SPE
SPE
23–n
NOISE FLOOR (SUBBAND 1)
NOISE FLOOR (SUBBAND n)
AVERAGE SPEECH POWER LEVEL (SUBBAND 1)
AVERAGE SPEECH POWER LEVEL (SUBBAND n)
24'
24–1
GC
GC
24–n
GAIN (SUBBAND 1)
GAIN (SUBBAND n)
25'
25–1
SB
SB
25–n
63
MODIFIED SPEECH SIGNAL